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(54) **NOISE SUPPRESSION ASSISTED AUTOMATIC SPEECH RECOGNITION**

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CPC ***G10L 21/02*** (2013.01); ***G10L 21/00*** (2013.01)

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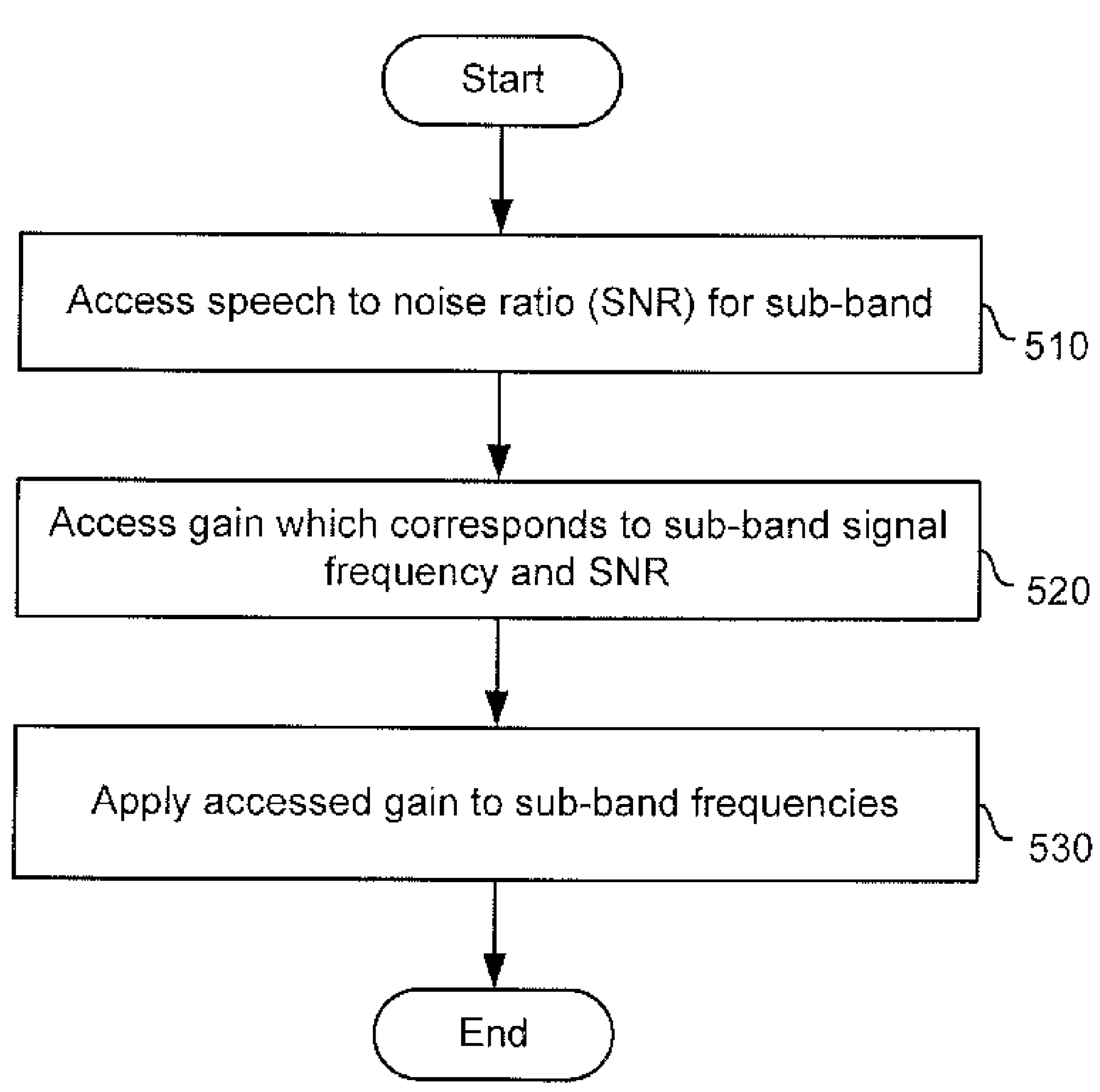
(57) **ABSTRACT**

Noise suppression information is used to optimize or improve automatic speech recognition performed for a signal. Noise suppression can be performed on a noisy speech signal using a gain value. The gain to apply to the noisy speech signal is selected to optimize speech recognition analysis of the resulting signal. The gain may be selected based on one or more features for a current sub band and time frame, as well as one or more features for other sub bands and/or time frames. Noise suppression information can be provided to a speech recognition module to improve the robustness of the speech recognition analysis. Noise suppression information can also be used to encode and identify speech.

14 Claims, 6 Drawing Sheets



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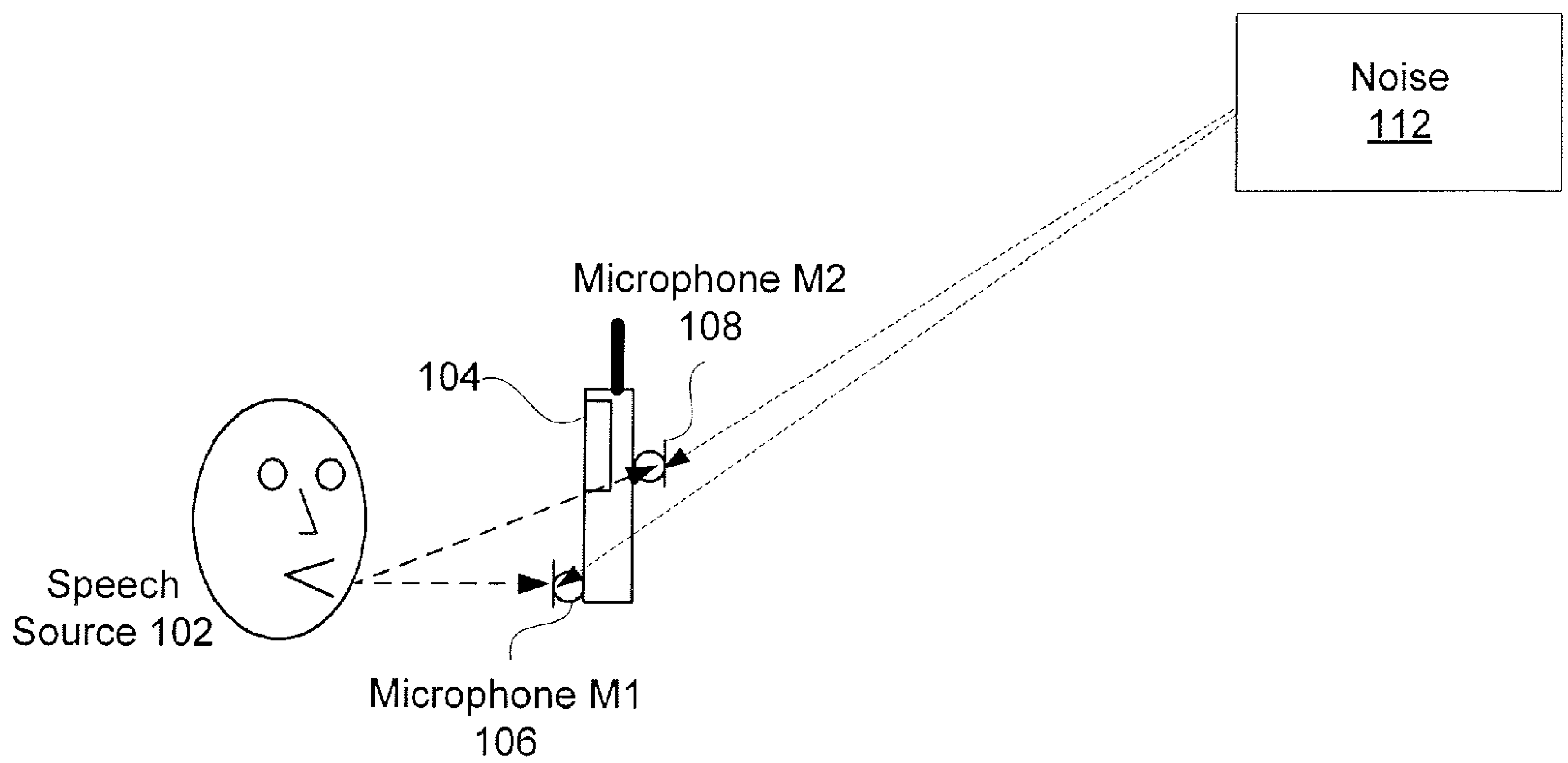

Noise
112
Microphone M2
108
104
Speech
Source 102
Microphone M1
106

FIGURE 1

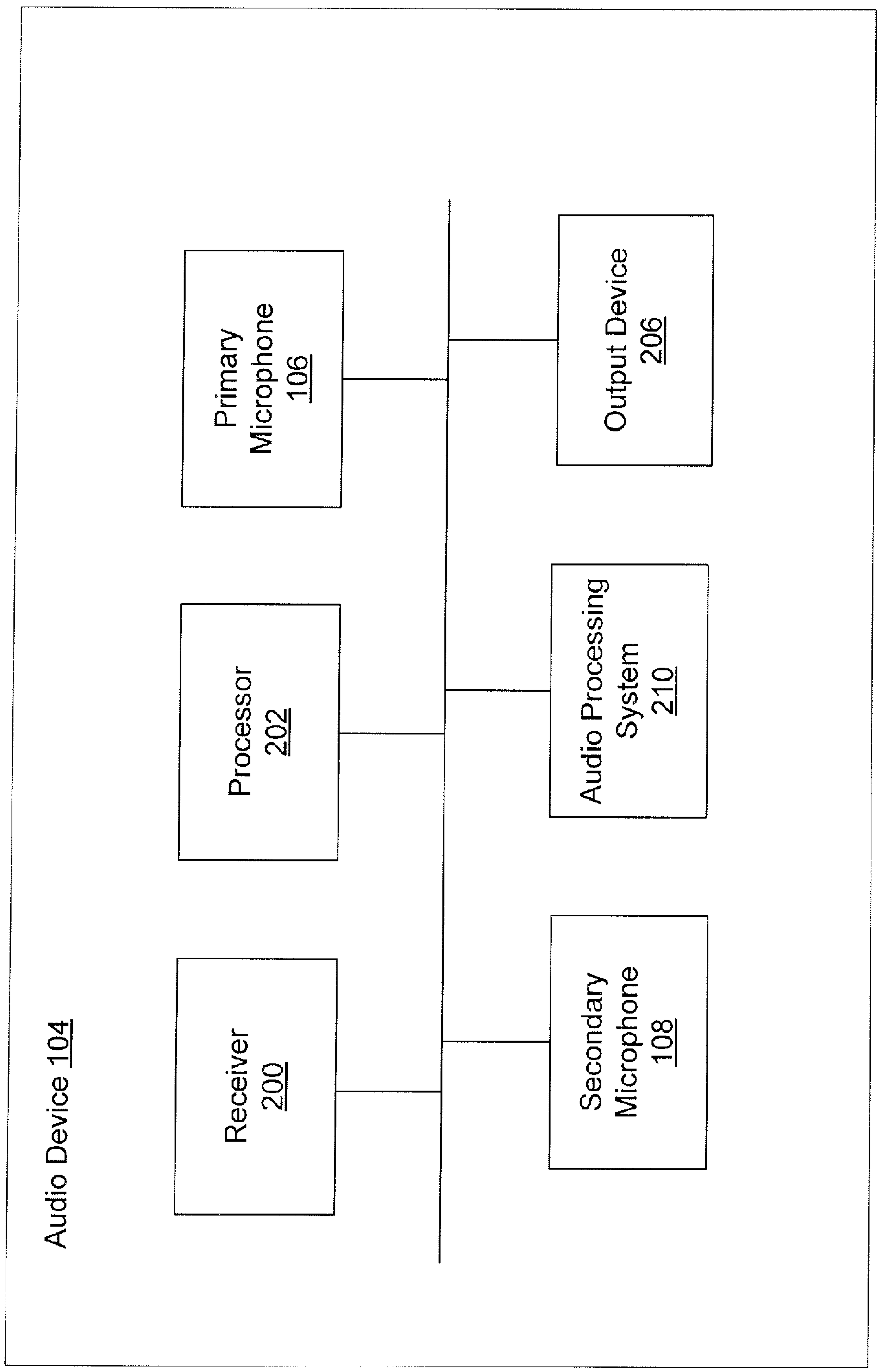
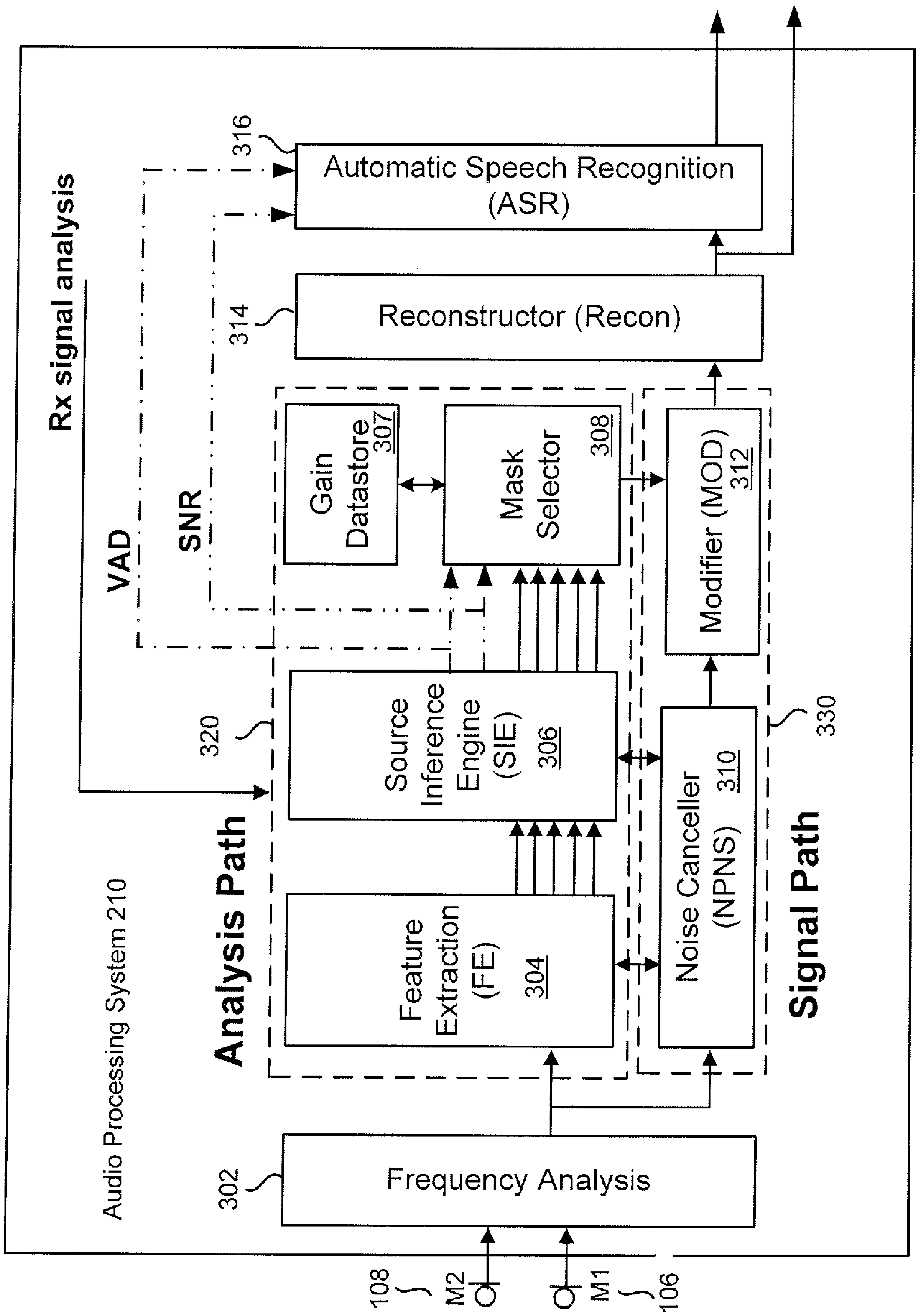

Audio Device 104
Receiver 200
Processor 202
Primary Microphone 106
Secondary Microphone 108
Audio Processing System 210
Output Device 206

FIGURE 2



Audio Processing System 210
Rx signal analysis
VAD
SNR
316
Automatic Speech Recognition (ASR)
314
Reconstructor (Recon)
Gain Datastore 307
Mask Selector 308
Modifier (MOD) 312
320
Source Inference Engine (SIE) 306
Noise Canceller (NPNS) 310
330
Analysis Path
Signal Path
Feature Extraction (FE) 304
302
Frequency Analysis
108
M2
M1
106

FIGURE 3

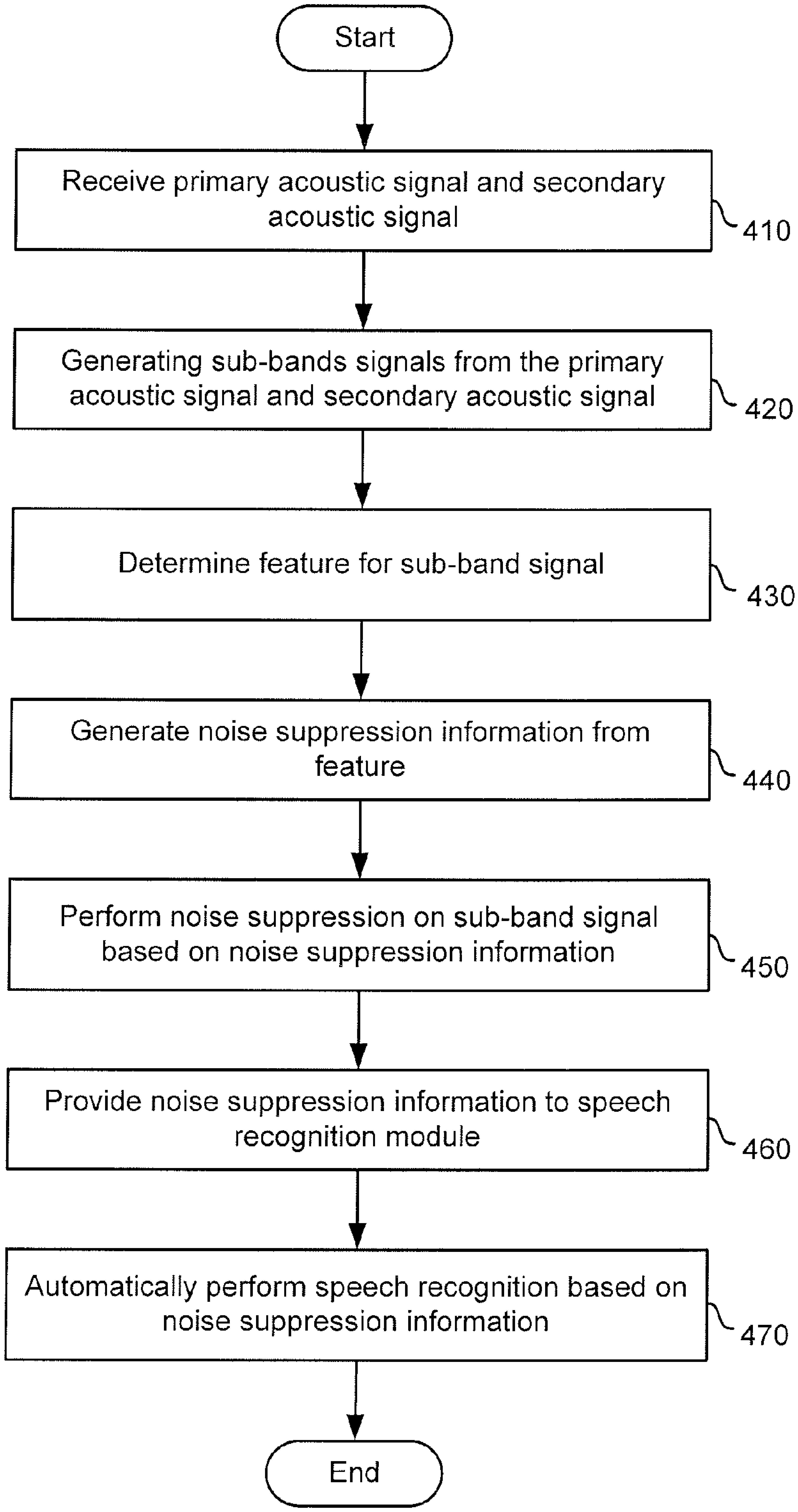

Start
Receive primary acoustic signal and secondary acoustic signal
410
Generating sub-bands signals from the primary acoustic signal and secondary acoustic signal
420
Determine feature for sub-band signal
430
Generate noise suppression information from feature
440
Perform noise suppression on sub-band signal based on noise suppression information
450
Provide noise suppression information to speech recognition module
460
Automatically perform speech recognition based on noise suppression information
470
End

FIGURE 4

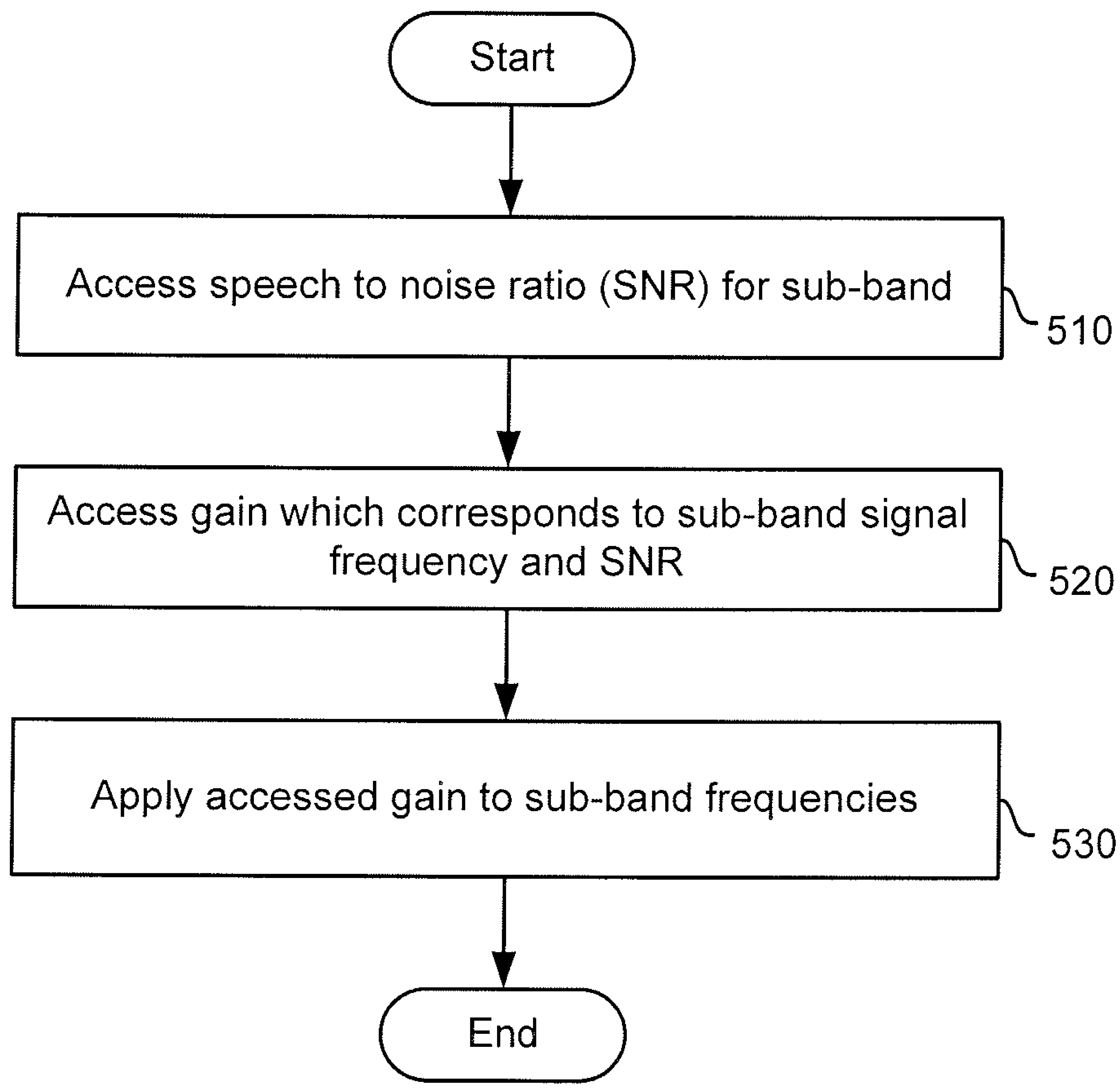

Start
Access speech to noise ratio (SNR) for sub-band
510
Access gain which corresponds to sub-band signal frequency and SNR
520
Apply accessed gain to sub-band frequencies
530
End

FIGURE 5

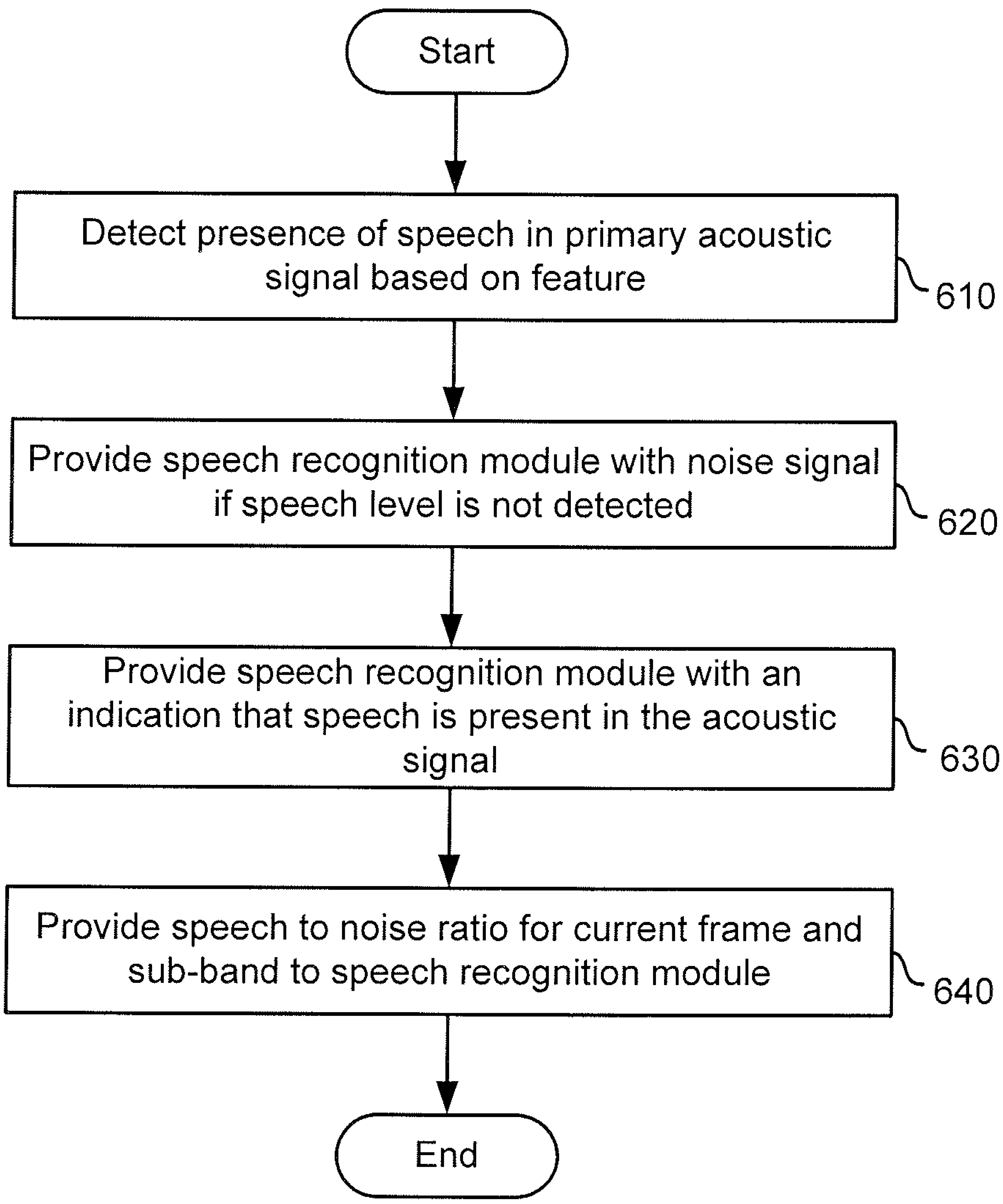

Start
Detect presence of speech in primary acoustic signal based on feature
610
Provide speech recognition module with noise signal if speech level is not detected
620
Provide speech recognition module with an indication that speech is present in the acoustic signal
630
Provide speech to noise ratio for current frame and sub-band to speech recognition module
640
End

FIGURE 6

NOISE SUPPRESSION ASSISTED AUTOMATIC SPEECH RECOGNITION

CROSS-REFERENCE TO RELATED APPLICATIONS

This application claims the priority benefit of U.S. Provisional Application Ser. No. 61/346,851, titled "Noise Suppression Assisted Automatic Speech Recognition," filed May 20, 2010, the disclosure of the aforementioned application is incorporated herein by reference.

BACKGROUND OF THE INVENTION

Speech recognition systems have been used to convert spoken words into text. In medium and high noise environments, however, the accuracy of automatic speech recognition systems tends to degrade significantly. As a result, most speech recognition systems are used with audio captured in a noise-free environment.

Unlike speech recognition systems, a standard noise reduction strategy consists of strongly attenuating portions of the acoustic spectrum which are dominated by noise. Spectrum portions dominated by speech are preserved.

Strong attenuation of undesired spectrum portions is a valid strategy from the point of view of noise reduction and perceived output signal quality, it is not necessarily a good strategy for an automatic speech recognition system. In particular, the spectral regions strongly attenuated by noise suppression may have been necessary to extract features for speech recognition. As a result, the attenuation resulting from noise suppression corrupts the features of the speech signal more than the original noise signal. This corruption by the noise suppression of the speech signal, which is greater than the corruption caused by the added noise signal, causes the noise reduction algorithm to make automatic speech recognition results unusable.

SUMMARY OF THE INVENTION

The present technology may utilize noise suppression information to optimize or improve automatic speech recognition performed for a signal. Noise suppression may be performed on a noisy speech signal using a gain value. The gain to apply to the noisy signal as part of the noise suppression is selected to optimize speech recognition analysis of the resulting signal. The gain may be selected based on one or more features for a current sub band and time frame, as well as others. Noise suppression information may be provided to a speech recognition module to improve the robustness of the speech recognition analysis. Noise suppression information may also be used to encode and identify speech. Resources spent on automatic speech recognition such as a bit rate of a speech codec) may be selected based on the SNR.

An embodiment may enable processing of an audio signal. Sub-band signals may be generated from a received primary acoustic signal and a secondary acoustic signal. One or more features may be determined for a sub-band signal. Noise suppression information may be determined based the one or more features to a speech recognition module.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. **1** illustrates an environment in which the present technology may be utilized.

FIG. **2** is a block diagram of an exemplary audio device.

FIG. **3** is a block diagram of an exemplary audio processing system.

FIG. **4** is a flow chart of an exemplary method for performing speech recognition based on noise suppression information.

FIG. **5** is a flow chart of an exemplary method for performing noise suppression on a sub band signal.

FIG. **6** is a flow chart of an exemplary method for providing noise suppression information to a speech recognition module.

DETAILED DESCRIPTION OF THE INVENTION

The present technology may utilize noise suppression information to optimize or improve automatic speech recognition performed for a signal. Noise suppression may be performed on a noisy speech signal using a gain value. The gain to apply to the noisy signal as part of the noise suppression is selected to optimize speech recognition analysis of the resulting signal. The gain may be selected based on one or more features for a current sub band and time frame, as well as others.

Noise suppression information may be provided to a speech recognition module to improve the robustness of the speech recognition analysis. Noise suppression information may include voice activity detection (VAD) information, such as for example noise, an indication of whether a signal includes speech, an indication of a speech to noise ration (SNR) for a signal, and other information. Noise suppression information may also be used to encode and identify speech. Resources spent on automatic speech recognition such as a bit rate of a speech codec) may be selected based on the SNR.

FIG. **1** is an illustration of an environment in which embodiments of the present technology may be used. A user may act as an audio (speech) source **102** to an audio device **104**. The exemplary audio device **104** includes two microphones: a primary microphone **106** relative to the audio source **102** and a secondary microphone **108** located a distance away from the primary microphone **106**. Alternatively, the audio device **104** may include a single microphone. In yet other embodiments, the audio device **104** may include more than two microphones, such as for example three, four, five, six, seven, eight, nine, ten or even more microphones.

The primary microphone **106** and secondary microphone **108** may be omni-directional microphones. Alternatively embodiments may utilize other forms of microphones or acoustic sensors, such as directional microphones.

While the microphones **106** and **108** receive sound (i.e. acoustic signals) from the audio source **102**, the microphones **106** and **108** also pick up noise **112**. Although the noise **110** is shown coming from a single location in FIG. **1**, the noise **110** may include any sounds from one or more locations that differ from the location of audio source **102**, and may include reverberations and echoes. The noise **110** may be stationary, non-stationary, and/or a combination of both stationary and non-stationary noise.

Some embodiments may utilize level differences (e.g. energy differences) between the acoustic signals received by the two microphones **106** and **108**. Because the primary microphone **106** is much closer to the audio source **102** than the secondary microphone **108** in a close-talk use case, the intensity level is higher for the primary microphone **106**, resulting in a larger energy level received by the primary microphone **106** during a speech/voice segment, for example.

The level difference may then be used to discriminate speech and noise in the time-frequency domain. Further embodiments may use a combination of energy level differences and time delays to discriminate speech. Based on binaural cue encoding, speech signal extraction or speech enhancement may be performed.

FIG. **2** is a block diagram of an exemplary audio device **104**. In the illustrated embodiment, the audio device **104** includes a receiver **200**, a processor **202**, the primary microphone **106**, an optional secondary microphone **108**, an audio processing system **210**, and an output device **206**. The audio device **104** may include further or other components necessary for audio device **104** operations. Similarly, the audio device **104** may include fewer components that perform similar or equivalent functions to those depicted in FIG. **2**.

Processor **202** may execute instructions and modules stored in a memory (not illustrated in FIG. **2**) in the audio device **104** to perform functionality described herein, including noise reduction for an acoustic signal, speech recognition, and other functionality. Processor **202** may include hardware and software implemented as a processing unit, which may process floating point operations and other operations for the processor **202**.

The exemplary receiver **200** may include an acoustic sensor configured to receive and transmit a signal to and from a communications network. In some embodiments, the receiver **200** may include an antenna device. The signal received may be forwarded to the audio processing system **210** to reduce noise using the techniques described herein, and provide an audio signal to the output device **206**. Similarly, a signal received by one or more of primary microphone **106** and secondary microphone **108** may be processed for noise suppression and ultimately transmitted to a communications network via receiver **200**. Hence, the present technology may be used in one or both of the transmit and receive paths of the audio device **104**.

The audio processing system **210** is configured to receive the acoustic signals from an acoustic source via the primary microphone **106** and secondary microphone **108** (or a far-end signal via receiver **200**) and process the acoustic signals. Processing may include performing noise reduction within an acoustic signal and speech recognition for an acoustic signal. The audio processing system **210** is discussed in more detail below.

The primary and secondary microphones **106**, **108** may be spaced a distance apart in order to allow for detecting an energy level difference, time difference or phase difference between them. The acoustic signals received by primary microphone **106** and secondary microphone **108** may be converted into electrical signals (i.e. a primary electrical signal and a secondary electrical signal). The electrical signals may themselves be converted by an analog-to-digital converter (not shown) into digital signals for processing in accordance with some embodiments. In order to differentiate the acoustic signals for clarity purposes, the acoustic signal received by the primary microphone **106** is herein referred to as the primary acoustic signal, while the acoustic signal received from by the secondary microphone **108** is herein referred to as the secondary acoustic signal. The primary acoustic signal and the secondary acoustic signal may be processed by the audio processing system **210** to produce a signal with an improved signal-to-noise ratio. It should be noted that embodiments of the technology described herein may be practiced utilizing only the primary microphone **106**.

The output device **206** is any device which provides an audio output to the user. For example, the output device **206** may include a speaker, an earpiece of a headset or handset, or a speaker on a conference device.

In various embodiments, where the primary and secondary microphones are omni-directional microphones that are closely-spaced (e.g., 1-2 cm apart), a beamforming technique may be used to simulate forwards-facing and backwards-facing directional microphones. The level difference may be used to discriminate speech and noise in the time-frequency domain which can be used in noise reduction.

FIG. **3** is a block diagram of an exemplary audio processing system **210** for performing noise reduction and automatic speech recognition. In exemplary embodiments, the audio processing system **210** is embodied within a memory device within audio device **104**. The audio processing system **210** may include a frequency analysis module **302**, a feature extraction module **304**, a source inference engine module **306**, gain data store **307**, mask selector module **308**, noise canceller module **310**, modifier module **312**, reconstructor module **314**, and automatic speech recognition **316**. Audio processing system **210** may include more or fewer components than illustrated in FIG. **3**, and the functionality of modules may be combined or expanded into fewer or additional modules. Exemplary lines of communication are illustrated between various modules of FIG. **3**, and in other figures herein. The lines of communication are not intended to limit which modules are communicatively coupled with others, nor are they intended to limit the number of and type of signals communicated between modules.

In operation, acoustic signals received from the primary microphone **106** and second microphone **108** are converted to electrical signals, and the electrical signals are processed through frequency analysis module **302**. The acoustic signals may be pre-processed in the time domain before being processed by frequency analysis module **302**. Time domain pre-processing may include applying input limiter gains, speech time stretching, and filtering using an FIR or IIR filter.

The frequency analysis module **302** receives the acoustic signals and mimics the frequency analysis of the cochlea (e.g., cochlear domain) to generate sub-band signals, simulated by a filter bank. The frequency analysis module **302** separates each of the primary and secondary acoustic signals into two or more frequency sub-band signals. A sub-band signal is the result of a filtering operation on an input signal, where the bandwidth of the filter is narrower than the bandwidth of the signal received by the frequency analysis module **302**. The filter bank may be implemented by a series of cascaded, complex-valued, first-order IIR filters. Alternatively, other filters such as short-time Fourier transform (STFT), sub-band filter banks, modulated complex lapped transforms, cochlear models, wavelets, etc., can be used for the frequency analysis and synthesis. The samples of the frequency sub-band signals may be grouped sequentially into time frames (e.g. over a predetermined period of time). For example, the length of a frame may be 4 ms, 8 ms, or some other length of time. In some embodiments there may be no frame at all. The results may include sub-band signals in a fast cochlea transform (FCT) domain.

The sub-band frame signals are provided from frequency analysis module **302** to an analysis path sub-system **320** and a signal path sub-system **330**. The analysis path sub-system **320** may process the signal to identify signal features, distinguish between speech components and noise components of the sub-band signals, and determine a signal modifier. The signal path sub-system **330** is responsible for modifying sub-band signals of the primary acoustic signal by reducing noise in the sub-band signals. Noise reduction can include applying a modifier, such as a multiplicative gain mask determined in the analysis path sub-system **320**, or by subtracting components from the sub-band signals. The noise reduction may reduce noise and preserve the desired speech components in the sub-band signals.

Signal path sub-system **330** includes noise canceller module **310** and modifier module **312**. Noise canceller module **310** receives sub-band frame signals from frequency analysis module **302**. Noise canceller module **310** may subtract (e.g., cancel) a noise component from one or more sub-band signals of the primary acoustic signal. As such, noise canceller module **310** may output sub-band estimates of noise components in the primary signal and sub-band estimates of speech components in the form of noise-subtracted sub-band signals.

Noise canceller module **310** may provide noise cancellation, for example in systems with two-microphone configurations, based on source location by means of a subtractive algorithm. Noise canceller module **310** may also provide echo cancellation and is intrinsically robust to loudspeaker and Rx path non-linearity. By performing noise and echo cancellation (e.g., subtracting components from a primary signal sub-band) with little or no voice quality degradation, noise canceller module **310** may increase the speech-to-noise ratio (SNR) in sub-band signals received from frequency analysis module **302** and provided to modifier module **312** and post filtering modules. The amount of noise cancellation performed may depend on the diffuseness of the noise source and the distance between microphones, both of which contribute towards the coherence of the noise between the microphones, with greater coherence resulting in better cancellation.

Noise canceller module **310** may be implemented in a variety of ways. In some embodiments, noise canceller module **310** may be implemented with a single NPNS module. Alternatively, Noise canceller module **310** may include two or more NPNS modules, which may be arranged for example in a cascaded fashion.

An example of noise cancellation performed in some embodiments by the noise canceller module **310** is disclosed in U.S. patent application Ser. No. 12/215,980, entitled "System and Method for Providing Noise Suppression Utilizing Null Processing Noise Subtraction," filed Jun. 30, 2008, U.S. application Ser. No. 12/422,917, entitled "Adaptive Noise Cancellation," filed Apr. 13, 2009, and U.S. application Ser. No. 12/693,998, entitled "Adaptive Noise Reduction Using Level Cues," filed Jan. 26, 2010, the disclosures of which are each incorporated herein by reference.

The feature extraction module **304** of the analysis path sub-system **320** receives the sub-band frame signals derived from the primary and secondary acoustic signals provided by frequency analysis module **302** as well as the output of NPNS module **310**. Feature extraction module **304** may compute frame energy estimations of the sub-band signals, inter-microphone level differences (ILD), inter-microphone time differences (ITD) and inter-microphones phase differences (IPD) between the primary acoustic signal and the secondary acoustic signal, self-noise estimates for the primary and second microphones, as well as other monaural or binaural features which may be utilized by other modules, such as pitch estimates and cross-correlations between microphone signals. The feature extraction module **304** may both provide inputs to and process outputs from NPNS module **310**.

NPNS module may provide noise cancelled sub-band signals to the ILD block in the feature extraction module **304**. Since the ILD may be determined as the ratio of the NPNS output signal energy to the secondary microphone energy, ILD is often interchangeable with Null Processing Inter-microphone Level Difference (NP-ILD). "Raw-ILD" may be used to disambiguate a case where the ILD is computed from the "raw" primary and secondary microphone signals.

Determining energy level estimates and inter-microphone level differences is discussed in more detail in U.S. patent application Ser. No. 11/343,524, entitled "System and Method for Utilizing Inter-Microphone Level Differences for Speech Enhancement", which is incorporated by reference herein.

Source inference engine module **306** may process the frame energy estimations provided by feature extraction module **304** to compute noise estimates and derive models of the noise and speech in the sub-band signals. Source inference engine module **306** adaptively estimates attributes of the acoustic sources, such as their energy spectra of the output signal of the NPNS module **310**. The energy spectra attribute may be utilized to generate a multiplicative mask in mask generator module **308**.

The source inference engine module **306** may receive the NP-ILD from feature extraction module **304** and track the NP-ILD probability distributions or "clusters" of the target audio source **102**, background noise and optionally echo.

This information is then used, along with other auditory cues, to define classification boundaries between source and noise classes. The NP-ILD distributions of speech, noise and echo may vary over time due to changing environmental conditions, movement of the audio device **104**, position of the hand and/or face of the user, other objects relative to the audio device **104**, and other factors. The cluster tracker adapts to the time-varying NP-ILDs of the speech or noise source(s).

An example of tracking clusters by a cluster tracker module is disclosed in U.S. patent application Ser. No. 12/004,897, entitled "System and method for Adaptive Classification of Audio Sources," filed on Dec. 21, 2007, the disclosure of which is incorporated herein by reference.

Source inference engine module **306** may include a noise estimate module which may receive a noise/speech classification control signal from the cluster tracker module and the output of noise canceller module **310** to estimate the noise N(t,w), wherein t is a point in time and W represents a frequency or sub-band. A speech to noise ratio (SNR) can be generated by source inference engine module **306** from the noise estimate and a speech estimate, and the SNR can be provided to other modules within the audio device, such as automatic speech recognition module **316** and mask selector **308**.

Gain data store **307** may include one or more stored gain values and may communicate with mask selector **308**. Each stored gain may be associated with a set of one or more features. An exemplary set of features may include a speech to noise ratio and a frequency (i.e., a center frequency for a sub band). Other feature data may also be stored in gain store **307**. Each gain stored in gain data store **307** may, when applied to a sub-band signal, provide as close to a clean speech signal as possible. Though the gains provide a speech signal with a reduced amount of noise, they may not provide the perceptually most desirable sounding speech.

In some embodiments, each gain stored in gain store **307** may be optimized for a set of features, such as for example a particular frequency and speech to noise ratio. For example, to determine an optimal gain value for a particular combination of features, a known speech spectrum may be combined with noise at various speech to noise ratios. Because the energy spectrum and noise are known, a gain can be determined which suppress the combined speech-noise signal into a clean speech signal which is ideal for speech recognition. In some embodiments, the gain is configured to suppress the speech-noise signal such that noise is reduced but no portion of the speech signal is attenuated or degraded. These gains derived from the combined signals for a known SNR are stored in the gain data store for different combination of frequency and speech to noise ratio.

Mask selector **308** may receive a set of one or more features and/or other data from source inference engine **306**, query gain data store **307** for a gain associated with a particular set of features and/or other data, and provide an accessed gain to modifier **312**. For example, for a particular sub band, mask selector **308** may receive a particular speech to noise ratio from source inference engine **306** for the particular sub band in the current frame. Mask selector **308** may then query data store **307** for a gain that is associated with the combination of the speech to noise ratio and the current sub band center frequency. Mask selector **308** receives the corresponding gain from gain data store **307** and provides the gain to modifier **312**.

The accessed gain may be applied to the estimated noise subtracted sub-band signals provided, for example as a multiplicative mask, by noise canceller **310** to modifier **312**. The modifier module **312** multiplies the gain masks to the noise-subtracted sub-band signals of the primary acoustic signal output by the noise canceller module **310**. Applying the mask reduces energy levels of noise components in the sub-band signals of the primary acoustic signal and results in noise reduction.

Modifier module **312** receives the signal path cochlear samples from noise canceller module **310** and applies a gain mask received from mask selector **308** to the received samples. The signal path cochlear samples may include the noise subtracted sub-band signals for the primary acoustic signal. The gain mask provided by mask selector **308** may vary quickly, such as from frame to frame, and noise and speech estimates may vary between frames. To help address the variance, the upwards and downwards temporal slew rates of the mask may be constrained to within reasonable limits by modifier **312**. The mask may be interpolated from the frame rate to the sample rate using simple linear interpolation, and applied to the sub-band signals by multiplicative noise suppression. Modifier module **312** may output masked frequency sub-band signals.

Reconstructor module **314** may convert the masked frequency sub-band signals from the cochlea domain back into the time domain. The conversion may include adding the masked frequency sub-band signals and phase shifted signals. Alternatively, the conversion may include multiplying the masked frequency sub-band signals with an inverse frequency of the cochlea channels. Once conversion to the time domain is completed, the synthesized acoustic signal may be output to the user via output device **206** and/or provided to a codec for encoding.

In some embodiments, additional post-processing of the synthesized time domain acoustic signal may be performed. For example, comfort noise generated by a comfort noise generator may be added to the synthesized acoustic signal prior to providing the signal to the user. Comfort noise may be a uniform constant noise that is not usually discernable to a listener (e.g., pink noise). This comfort noise may be added to the synthesized acoustic signal to enforce a threshold of audibility and to mask low-level non-stationary output noise components. In some embodiments, the comfort noise level may be chosen to be just above a threshold of audibility and may be settable by a user. In some embodiments, the mask generator module **308** may have access to the level of comfort noise in order to generate gain masks that will suppress the noise to a level at or below the comfort noise.

Automatic speech recognition module **316** may perform a speech recognition analysis on the reconstructed signal output by reconstructor **314**. Automatic speech recognition module **316** may receive a voice activity detection (VAD) signal as well as a speech to noise (SNR) ratio indication or other noise suppression information from source inference engine **306**. The information received from source information engine **306**, such as the VAD and SNR, may be used to optimize the speech recognition process performed by automatic speech recognition module **316**. Speech recognition module **316** is discussed in more detail below.

The system of FIG. **3** may process several types of signals received by an audio device. The system may be applied to acoustic signals received via one or more microphones. The system may also process signals, such as a digital Rx signal, received through an antenna or other connection.

An exemplary system which may be used to implement at least a portion of audio processing system **210** is described in U.S. patent application Ser. No. 12/832,920, titled "Multi-Microphone Robust Noise Suppression," filed Jul. 8, 2010, the disclosure of which is incorporated herein by reference

FIG. **4** is a flow chart of an exemplary method for performing speech recognition based on noise suppression information. First, a primary acoustic signal and a secondary acoustic signal are received at step **410**. The signals may be received through microphones **106** and **108** of audio device **104**. Sub band signals may then be generated from the primary acoustic signal and secondary acoustic signal at step **420**. The received signals may be converted to sub band signals by frequency analysis module **302**.

A feature is determined for a sub band signal at step **430**. Feature extractor **304** may extract features for each sub band in the current frame or the frame as a whole. Features may include a speech energy level for a particular sub band noise level, pitch, and other features. Noise suppression information is then generated from the features at step **440**. The noise suppression information may be generated and output by source inference engine **306** from features received from feature extraction module **304**. The noise suppression information may include an SNR ratio for each sub band in the current frame, a VAD signal for the current frame, ILD, and other noise suppression information.

Noise suppression may be performed on a sub band signal based on noise suppression information at step **450**. The noise suppression may include accessing a gain value based on one or more features and applying the gain to a sub band acoustic signal. Performing noise suppression on a sub band signal is discussed in more detail below with respect to the method of FIG. **5**. Additionally, noise suppression performed on a sub band signal may include performing noise cancellation by noise canceller **310** in the audio processing system of FIG. **3**.

Noise suppression information may be provided to speech recognition module **316** at step **460**. Speech recognition module **316** may receive noise suppression information to assist with speech recognition. Providing noise suppression information to speech recognition module **316** is discussed in more detail below with respect to FIG. **6**.

Speech recognition is automatically performed based on the noise suppression information at step **470**. The speech recognition process may be optimized based on the noise suppression information. Performing speech recognition based on noise suppression information may include modulating a bit rate of a speech encoder or decoder based on a speech to noise ratio for a particular frame. In some embodiments, the bit rate is decreased when the speech to noise ratio is large. In some embodiments, speech recognition based on noise suppression may include setting a node search depth level by a speech recognition module based on a speech to noise ratio for a current frame. The node search depth level, for example, may be decreased when the speech to noise ratio is large.

FIG. **5** is a flow chart of an exemplary method for performing noise suppression on a sub band signal. The method of FIG. **5** provides more detail for step **450** in the method of FIG. **4**. A speech to noise ratio (SNR) for a sub band is accessed at step **510**. The SNR may be received by mask selector **308** from source inference engine **306**. Mask selector **308** also has access to sub band information for the sub band being considered.

A gain which corresponds to the sub band signal frequency and the signal of the noise ratio is accessed at step **520**. The gain is accessed by mask selector **308** from gain data store **307** and may correspond to a particular sub band signal frequency and SNR. The accessed gain is then applied to one or more sub band frequencies at step **530**. The accessed gain may be provided to modifier **312** which then applies the gain to a sub band which may or may not be have undergone noise cancellation.

FIG. **6** is a flow chart of an exemplary method for providing noise suppression information to a speech recognition module. The method of FIG. **6** may provide more detail for step **460** than the method of FIG. **4**. A determination as to whether speech is detected in a primary acoustic signal based on one or more features is performed at step **610**. The detection may include detecting whether speech is or is not present within the signal within the current frame. In some embodiments, an SNR for the current sub band or for all sub bands may be compared to a threshold level. If the SNR is above the threshold value, then speech may be detected to be present in the primary acoustic signal. If the SNR is not above the threshold value, then speech may be determined to not be present in the current frame.

Each of steps **620**-**640** describe how speech recognition may be optimized based on noise suppression or noise suppression information and may be performed in combination or separately. Hence, in some embodiments, only one of step **620**-**640** may be performed. In some embodiments, more than just one of steps **620**-**640** may be performed when providing noise suppression information to a speech recognition module.

A speech recognition module is provided with a noise signal if a speech is not detected in a current frame of a signal at step **620**. For example, if the determination is made that speech is not present in the current frame of a reconstructed signal, a noise signal is provided to acoustic speech recognition module **316** in order to ensure that no false positive for speech detection occurs. The noise signal may be any type of signal that has a high likelihood of not being mistaken for speech by the speech recognition module.

A speech recognition module may be provided with an indication that speech is present in the acoustic signal at step **630**. In this case, automatic speech recognition module **316** may be provided with a VAD signal provided by source inference engine **306**. The VAD signal may indicate whether or not speech is present in the signal provided to automatic speech recognition module **316**. Automatic speech recognition module **316** may use the VAD signal to determine whether or not to perform speech recognition on the signal.

A speech to noise ratio (SNR) signal may be provided for the current frame and/or sub band to the speech recognition module at step **640**. In this case, the SNR may provide a value within a range of values indicating whether or not speech is present. This may help the automatic speech recognition module learn when to expend resources to recognize speech and when not to.

The above described modules, including those discussed with respect to FIG. **3**, may include instructions stored in a storage media such as a machine readable medium (e.g., computer readable medium). These instructions may be retrieved and executed by the processor **202** to perform the functionality discussed herein. Some examples of instructions include software, program code, and firmware. Some examples of storage media include memory devices and integrated circuits.

While the present invention is disclosed by reference to the preferred embodiments and examples detailed above, it is to be understood that these examples are intended in an illustrative rather than a limiting sense. It is contemplated that modifications and combinations will readily occur to those skilled in the art, which modifications and combinations will be within the spirit of the invention and the scope of the following claims.

What is claimed is:

1. A method for processing an audio signal, comprising:

generating sub-band signals from a received primary acoustic signal and a received secondary acoustic signal;

determining two or more features for the sub-band signals, the two or more features including a speech energy level for the sub-band noise level and at least one of the following: inter-microphone level differences, inter-microphone time differences, and inter-microphone phase differences between the primary acoustic signal and the secondary acoustic signal;

suppressing a noise component in the primary acoustic signal based on the two or more features, the suppressing configured to clean the primary acoustic signal to create a cleaned speech signal optimized for accurate speech recognition processing by an automatic speech recognition processing module, the suppressing comprising:

applying a gain to a sub-band of the primary acoustic signal to provide a noise suppressed signal, the applying comprising:

determining a speech to noise ratio (SNR) for the sub-band of the primary acoustic signal;

accessing the gain, based on the frequency of the sub-band and the determined SNR for the sub-band, from a datastore, the datastore including a plurality of pre-stored gains configured to create cleaned speech signals optimized for accurate speech recognition processing by the automatic speech recognition processing module, each pre-stored gain in the plurality of pre-stored gains associated with a corresponding frequency and an SNR value; and

applying the accessed gain to the sub-band frequency; and providing the cleaned speech signal and corresponding noise suppression information to the automatic speech recognition processing module, the noise suppression information based on the two or more features and including a voice activity detection signal.

2. The method of claim **1**, further comprising determining whether the primary acoustic signal includes speech, the determination performed based on the two or more features.

3. The method of claim **2**, further comprising providing a noise signal to the automatic speech recognition processing module in response to detecting that the primary acoustic signal does not include speech.

4. The method of claim **2**, wherein the voice activity detection signal is generated based on the determination of whether the primary acoustic signal includes speech, and the voice activity detection signal indicating whether automatic speech recognition is to occur.

5. The method of claim **4**, wherein the voice activity detection signal is a value within a range of values corresponding to the level of speech detected in the primary acoustic signal.

6. The method of claim **2**, wherein the noise suppression information includes a speech to noise ratio for the current time frame and the sub-band to the automatic speech recognition processing module.

7. The method of claim **1**, wherein the noise suppression information includes a speech to noise ratio, the method further comprising modulating a bit rate of a speech encoder or decoder based on the speech to noise ratio for a particular frame.

8. The method of claim **1**, wherein the noise suppression information includes a speech to noise ratio, the method further comprising setting a node search depth level by the automatic speech recognition processing module based on the speech to noise ratio for a current frame.

9. A non-transitory computer readable storage medium having embodied thereon a program, the program being executable by a processor to perform a method for reducing noise in an audio signal, the method comprising:

generating sub-band signals from a received primary acoustic signal and a received secondary acoustic signal;

determining two or more features for a sub-band signal, the two or more features including a speech energy level for the sub-band noise level and at least one of the following: inter-microphone level differences, inter-microphone time differences, and inter-microphone phase differences between the primary acoustic signal and the secondary acoustic signal;

suppressing a noise component in the primary acoustic signal based on the two or more features, the suppressing configured to clean the primary acoustic signal to create a cleaned speech signal optimized for accurate speech recognition processing by an automatic speech recognition processing module, the suppressing comprising:

applying a gain to a sub-band of the primary acoustic signal to provide a noise suppressed signal, the applying comprising:

determining a speech to noise ratio (SNR) for the sub-band of the primary acoustic signal;

accessing the gain, based on the frequency of the sub-band and the determined SNR for the sub-band, from a datastore, the datastore including a plurality of pre-stored gains configured to create cleaned speech signals optimized for accurate speech recognition processing by the automatic speech recognition processing module, each pre-stored gain in the plurality of pre-stored gains associated with a corresponding frequency and an SNR value; and

applying the accessed gain to the sub-band frequency; and

providing the cleaned speech signal and corresponding noise suppression information to the automatic speech recognition processing module, the noise suppression information based on the two or more features and including a speech to noise ratio for each of the sub-band signals and a voice activity detection signal.

10. The non-transitory computer readable storage medium of claim **9**, further comprising providing a noise signal to the automatic speech recognition processing module in response to detecting that the primary acoustic signal does not include speech.

11. The non-transitory computer readable storage medium of claim **9**, wherein the voice activity detection signal is generated based on the determination of whether the primary acoustic signal includes speech, and the voice activity detection signal indicating whether automatic speech recognition is to occur.

12. The non-transitory computer readable storage medium of claim **9**, wherein the noise suppression information includes a speech to noise ratio for the current time frame and the sub-band to the automatic speech recognition processing module.

13. The non-transitory computer readable storage medium of claim **9**, wherein the noise suppression information includes a speech to noise ratio, the method further comprising modulating a bit rate of a speech encoder or decoder based on the speech to noise ratio for a particular frame.

14. The non-transitory computer readable storage medium of claim **9**, wherein the noise suppression information includes a speech to noise ratio, the method further comprising setting a node search depth level by the automatic speech recognition processing module based on the speech to noise ratio for a current frame.

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